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10/533,612	04/29/2005	Kohei Asada	SONYJP 3.3-1024	6316

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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

PTOL-90A (Rev. 04/07)

Office Action Summary	Application No.	Applicant(s)
	10/533,612	ASADA ET AL.
	Examiner	Art Unit
	Joseph Saunders	2615

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 25 May 2005.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-20 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-20 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☒ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 29 April 2005 is/are: a)☐ accepted or b)☒ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☒ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date 7-3-06, 9-18-06.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

1. This is the initial office action based on the communications filed May 25, 2005. Claims 1 – 20 are currently pending and considered below.

Priority

2. Acknowledgment is made of applicant's claim for foreign priority based on an application filed in Japan on November 15, 2002. It is noted, however, that applicant has not filed a certified copy of the JP 2002-332565 application as required by 35 U.S.C. 119(b).

Drawings

3. Figures 1 – 3, 5, 6, and 15 should be designated by a legend such as --Prior Art-- because only that which is old is illustrated. See MPEP § 608.02(g). Corrected drawings in compliance with 37 CFR 1.121(d) are required in reply to the Office action to avoid abandonment of the application. The replacement sheet(s) should be labeled "Replacement Sheet" in the page header (as per 37 CFR 1.84(c)) so as not to obstruct any portion of the drawing figures. If the changes are not accepted by the examiner, the applicant will be notified and informed of any required corrective action in the next Office action. The objection to the drawings will not be held in abeyance.

Specification

4. The disclosure is objected to because of the following informalities: In the specification the delay time is referred to as the symbol pi (π) and should be corrected to the symbol tau (τ) as disclosed in the drawings.

Appropriate correction is required.

5. The lengthy specification has not been checked to the extent necessary to determine the presence of all possible minor errors. Applicant's cooperation is requested in correcting any errors of which applicant may become aware in the specification.

Double Patenting

6. A rejection based on double patenting of the "same invention" type finds its support in the language of 35 U.S.C. 101 which states that "whoever invents or discovers any new and useful process ... may obtain a patent therefor ..." (Emphasis added). Thus, the term "same invention," in this context, means an invention drawn to identical subject matter. See *Miller v. Eagle Mfg. Co.*, 151 U.S. 186 (1894); *In re Ockert*, 245 F.2d 467, 114 USPQ 330 (CCPA 1957); and *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970).

A statutory type (35 U.S.C. 101) double patenting rejection can be overcome by canceling or amending the conflicting claims so they are no longer coextensive in scope. The filing of a terminal disclaimer cannot overcome a double patenting rejection based upon 35 U.S.C. 101.

7. Claims 1, 2, 10, and 11 are provisionally rejected under 35 U.S.C. 101 as claiming the same invention as that of claim 2 and 4 of copending Application No.10/706,772. This is a provisional double patenting rejection since the conflicting claims have not in fact been patented.

Claim 1: Application No.10/706,772 discloses in claim 2 an audio signal processing method comprising the steps of ("method of reproducing an audio signal"): supplying an audio signal to each of a plurality of digital filters ("supplying an audio signal to a plurality of digital filters, respectively"); respectively supplying outputs from the plurality of digital filters to a plurality of speakers arranged in a speaker array to form a sound field ("generating a sound field inside a closed space by supplying respective outputs of the plurality of digital filters to a plurality of speakers constituting a speaker array, respectively"); setting a predetermined delay time in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with each other ("supplying the sounds outputted from the speaker array to a location of a listener inside the sound filed after being reflected by a wall surface of the closed space with a sound pressure larger than that of a peripheral location by setting predetermined delay times for said plurality of digital filters, respectively"); and adjusting an amplitude characteristic of the plurality of digital filters so that a low-pass filter characteristic will be given to a synthesis response of the audio signal at a second point in the sound field ("a sound pressure directly arriving at said listener from said speaker array is reduced by setting predetermined amplitudes to said plurality of digital filters, respectively").

Claim 2: Application No.10/706,772 further discloses in claim 2 the audio signal processing method according to claim 1, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface ("reflected by a wall surface").

Claims 10 and 11: Application No.10/706,772 also discloses in claim 4 an apparatus performing the method disclosed above and therefore, claim 4 also discloses the same invention as claims 10 and 11.

Claim Rejections - 35 USC § 102

(b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

8. Claims 1 – 4 and 10 – 13 are rejected under 35 U.S.C. 102(b) as being anticipated by Bienek et al. (WO 02/078388 A2), hereinafter Bienek.

Claim 1: Bienek discloses an audio signal processing method (method and apparatus to create a sound field) comprising the steps of: supplying an audio signal (input signal 101) to each of a plurality of digital filters (delay means 1508 or adjustable digital filter 1512 can also be arranged to apply delays); respectively supplying outputs from the plurality of digital filters to a plurality of speakers arranged in a speaker array to form a sound field (Description of Figure 6, Pages 18 – 19); setting a predetermined delay time in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital

filters and each of the plurality of speakers will coincide with each other (Third Sound Field, Pages 21 – 22 and Figure 7C and Figure 8); and adjusting an amplitude characteristic of the plurality of digital filters so that a low-pass filter characteristic will be given to a synthesis response of the audio signal at a second point in the sound field (Third Aspect of the Invention, Pages 26 – 27 and Figure 11).

Claim 2: Bienek discloses the audio signal processing method according to claim 1, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface (Figure 8).

Claim 3: Bienek discloses the audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the plurality of digital filters (Third Sound Field, Pages 21 – 22 and Figure 7C).

Claim 4: Bienek discloses the audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters (stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27).

Claim 10: Bienek discloses an audio signal processor (method and apparatus to create a sound field) comprising a plurality of digital filters (delay means 1508 or adjustable digital filter can also be arranged to apply delays) each supplied with an audio signal (input signal 101), wherein each of the plurality of digital filters supplies an output signal to each of a plurality of speakers arranged in a speaker array to form a sound field (Description of Figure 6, Pages 18 – 19); each of the plurality of digital filters has a predetermined delay time so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with each other (Third Sound Field, Pages 21 – 22 and Figure 7C and Figure 8); and each of the plurality of digital filters has an amplitude characteristic so that a low-pass filter characteristic will be given to a synthesis response of the audio signal at a second point in the sound field (Third Aspect of the Invention, Pages 26 – 27 and Figure 11).

Claim 11: Bienek discloses the audio signal processor according to claim 10, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface (Figure 8).

Claim 12: Bienek discloses the audio signal processor according to claim 10, wherein when forming the first and second points in the sound filter, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the plurality of digital filters (Third Sound Field, Pages 21 – 22 and Figure 7C).

Claim 13: Bienek discloses the audio signal processor according to claim 10, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters (stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27).

Claim Rejections - 35 USC § 103

9. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

10. Claims 5 – 9 and 14 – 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Bienek in view of Masako et al. (JP-8-191225-A), hereinafter Masako.

Claim 5: Bienek discloses the audio signal processing method according to claim 1, but does not explicitly disclose wherein: the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal; over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than a sampling period to provide a sample train and, wherein the

sample train is down-sampled to provide pulse-waveform data of the sampling period; and factor data is set for a part to be delayed by the plurality of digital filters based on the pulse- waveform data. Bienek does disclose "the minimum delay possible for a given signal is preferably as small or smaller than Ts, that signal's sample period" and that "most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sampling period. Otherwise, interpolation of the signal is necessary," Page 12 Lines 17 – 24. Therefore Bienek does disclose a fractional delay and also discloses that a delay filter and an adaptive digital filter may be used. Bienek does not disclose details of how to perform, for example, the interpolation necessary for the fractional delays disclosed above and therefore one would be inclined to look elsewhere for such a teaching. Masako discloses the technique necessary to include a fractional delay (Figure 6 –7 and Paragraph 29 – 33) and discloses that this technique is an effective approach when slight spacing differences influence the felling of the direction of perceived sound. Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to include the process of over-sampling an impulse response and then down-sampling to provide a pulse-waveform data of the sampling period as disclosed by Masako for setting the delay coefficients for the plurality of digital filters as disclosed by Bienek, thereby providing better perceived spatial resolution.

Claim 6: Bienek and Masako disclose the audio signal processing method according to claim 5, wherein the audio signal is delayed by a part of the predetermined delay time,

which is a multiple of the sampling period, by digital delay circuits which operate for the sampling period, while it is being delayed by the remainder of the predetermined delay time, which includes the decimal part by the digital filters (Bienek discloses that the delay time may be a fractional sampling period and also discloses cascading a delay means with adjustable digital filter means that can also apply delays. Therefore given the disclosure of Bienek and the teachings of Masako of how to calculate a finer representation of an impulse response, Bienek and Masako disclose implementing a delay using a simple delay element and an adjustable digital filter for the remainder or fractional part of the delay in a two stage process as disclosed by Bienek).

Claim 7: Bienek and Masako disclose the audio signal processing method according to claim 5, and wherein: an over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part (Masako, Paragraph 26).

Claim 8: Bienek and Masako disclose the audio signal processing method according to claim 7, wherein: the pulse-waveform data to be delayed by a delay time which is m/N (m = 1 to N - 1) of the sampling period is pre-stored in a data base; and pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters (Bienek, stored sets of delays

(for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27).

Claim 9: Bienek and Masako disclose the audio signal processing method according to claim 5, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data and set as a filter factor of each of the plurality of digital filters ("convolution multiplier", Masako, Paragraph 26).

Claim 14: Bienek discloses the audio signal processor according to claim 10, wherein: the pulse-waveform provided by the calculation circuit is set as a filter factor of each of the plurality of digital filters (Third Sound Field, Pages 21 – 22 and Figure 7C) but does not explicitly disclose the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal, there is further provided a calculation circuit to calculate pulse-waveform data of the sampling period by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train. Bienek does disclose "the minimum delay possible for a given signal is preferably as small or smaller than Ts, that signal's sample period" and that "most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sampling period. Otherwise, interpolation of the signal is necessary," Page 12 Lines 17 – 24. Therefore Bienek does

disclose a fractional delay and also discloses that a delay filter and an adaptive digital filter may be used. Bienek does not disclose details of how to perform, for example, the interpolation necessary for the fractional delays disclosed above and therefore one would be inclined to look elsewhere for such a teaching. Masako discloses the technique necessary to include a fractional delay (Figure 6 –7 and Paragraph 29 – 33) and discloses that this technique is an effective approach when slight spacing differences influence the felling of the direction of perceived sound. Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to include the process of over-sampling an impulse response and then down-sampling to provide a pulse-waveform data of the sampling period as disclosed by Masako for setting the delay coefficients for the plurality of digital filters as disclosed by Bienek, thereby providing better perceived spatial resolution.

Claim 15: Bienek and Masako disclose the audio signal processor according to claim 14, wherein: an over-sampling period of the over-sampling in the calculation circuit is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part (Masako, Paragraph 26).

Claim 16: Bienek and Masako disclose the audio signal processor according to claim 14, wherein a transfer characteristic providing a predetermined acoustic effect is

convoluted in the pulse-waveform data to set synthetic-waveform data as a filter factor of each of the plurality of digital filters ("convolution multiplier", Masako, Paragraph 26).

Claim 17: Bienek discloses the audio signal processor according to claim 10, wherein: the pulse-waveform data stored in the storing means is taken out and set as a filter factor of each of the plurality of digital filters (stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27) but does not explicitly disclose the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio Signal; there is further provided a storing means for storing pulse-waveform data of the sampling period provided by over- sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train. Bienek does disclose "the minimum delay possible for a given signal is preferably as small or smaller than Ts, that signal's sample period" and that "most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sampling period. Otherwise, interpolation of the signal is necessary," Page 12 Lines 17 – 24. Therefore Bienek does disclose a fractional delay and also discloses that a delay filter and an adaptive digital filter may be used. Bienek does not disclose details of how to perform, for example, the interpolation necessary for the fractional delays disclosed above and therefore one would be inclined to look elsewhere for such a teaching. Masako discloses the technique necessary to

include a fractional delay (Figure 6 –7 and Paragraph 29 – 33) and discloses that this technique is an effective approach when slight spacing differences influence the felling of the direction of perceived sound. Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to include the process of over-sampling an impulse response and then down-sampling to provide a pulse-waveform data of the sampling period as disclosed by Masako for setting the delay coefficients for the plurality of digital filters as disclosed by Bienek, thereby providing better perceived spatial resolution.

Claim 18: Bienek and Masako disclose the audio signal processor according to claim 17, wherein: an over-sampling period of the over-sampling is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part (Masako, Paragraph 26).

Claim 19: Bienek and Masako disclose the audio signal processor according to claim 17, wherein: a plurality of the pulse-waveform data corresponding to the decimal part is pre-stored in the storing means; and pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters (Bienek, stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27).

Claim 20: Bienek and Masako disclose the audio signal processor according to claim 17, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set the pulse-waveform data as a filter factor of each of the plurality of digital filters ("convolution multiplier", Masako, Paragraph 26).

Conclusion

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Joseph Saunders whose telephone number is (571) 270-1063. The examiner can normally be reached on Monday - Thursday, 9:00 a.m. - 4:00 p.m., EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Sinh Tran can be reached on (571) 272-7564. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

JS
September 28, 2007

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SUPERVISORY PATENT EXAMINER